QoE in multi-service multi-agent networks

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Abstract: The growth of emerging multimedia applications with different quality requirements has made the objective network QoS (NQoS) to be commonly considered as an inadequate metric for quality assessment in telecommunication services, being the quality of experience (QoE) concept the most predominant alternative. However, most QoE-related research studies are service-specific, not easily expandable to other multimedia services and unsuitable for tuning up network management procedures. Instead, we propose an integrated QoE management model and a calculus process that estimates final user-satisfaction over multi-service scenarios and identifies those elements that have a greater impact on users' QoE. In order to evaluate this impact, new VoIP and video perception assessment methods were developed, while state-of-the-art methods were used for web and online game services. The effectiveness of the model is demonstrated by applying a stepwise multiple linear regression method to identify both subjective and objective bottlenecks in the network.

Keywords: QoS; network QoS; NQoS; perceived QoS; PQoS; quality of experience; QoE; mean opinion score; MOS; analytic hierarchy process; AHP.

Reference to this paper should be made as follows: Liberal, F., Ferro, A., Koumaras, H., Kourtis, A., Sun, L. and Ifeachor, E.C. (2010) 'QoE in multi-service multi-agent networks', *Int. J. Communication Networks and Distributed Systems*, Vol. 4, No. 2, pp.183–206.

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1 Introduction

Quality of experience (QoE) or perceived QoS (PQoS) deals with assessing, quantifying and managing overall user-satisfaction regarding multimedia services. However, the concept of QoE is significantly differentiated by the type of the delivered service, namely video, voice over IP (VoIP) and online gaming services. More specifically, multimedia applications use video encoding techniques (e.g., MPEG-4/H.264) in order to achieve high compression by exploiting the spatial and temporal (S-T) redundancy in the original uncompressed video sequence. This procedure along with possible transmission errors during the streaming of the compressed data cause image artefacts, which in turn results in perceived quality degradation. Considering that the parameters with strong influence on the video quality are normally those set at the encoder (the most important being the bit rate, the frame rate and the resolution), the issue of evaluating QoE for video encoded services, considering flawless video transmission, is expressed in terms of encoding video quality in correlation with the encoding parameters (e.g., codec type, picture resolution, bit rate, frame rate, etc.).

Similarly, for VoIP applications, the QoE is affected by equivalent application level parameters (e.g., codec type, FEC, loss concealment and playout buffer algorithms) and network-related impairments including packet loss, delay and delay variation.

Furthermore, as new and emerging game consoles include out-of-the-box communication facilities, online gaming is becoming one of the most popular QoS demanding services, making online-gamers nearly the most QoS-conscious group of users.

Thus, there is a specialised QoE approach for each multimedia service (e.g., video metrics for MPEG-coded services and e-model for voice applications). At the same time, it is crucial for content/service providers and network operators to assess, predict and possibly control the end-to-end experienced quality for both commercial and technical reasons. Then, considering the differences between QoE estimation methods for different services, the need for a common multimedia description framework (i.e., MPEG-21) to help move toward an integrated provision of the various multimedia services over heterogeneous networks and terminal devices is widely recognised.

On the other hand, emerging communication services and convergence scenarios [as e.g., in IP multimedia subsystem (IMS) scenarios] will involve users who have simultaneously access to diverse multimedia services (video/voice/data). Thus, the barriers between the various types of services drop and the need for a generic QoE model, which will be capable of providing a single index of perceived quality for several services becomes obvious.

In this context, this paper proposes a generic QoE evaluation framework, which unifies various specialised perceived quality assessment methods. The proposed model allows the evaluation of the QoE achieved by the composition of diverse multimedia services and provides an overall picture of the user satisfaction, taking under consideration not only technical aspects of the services, but also subjective aspects, which usually have greater impact on user satisfaction than network QoS (NQoS).

Upon this introductory section, the rest of the paper is organised as follows: in Section 2, it presents a review on relative existing research regarding integrating QoE for different services (the so-called integrated approaches). In Section 3, the proposed generic QoE model is described, including a calculus process in order to estimate user-satisfaction. In Section 4, we review the QoE vs. NQoS mapping methods used in the calculus process. Later, in Section 5, the efficiency of the proposed model is examined to a case study: Evaluating different contributions of each agent to final satisfaction in a multi-service/agent scenario. Finally, the conclusions and possible future perspectives are discussed in the last section of this paper.

2 Background of integrated approaches

QoS is explicitly defined by ITU-T E.800 recommendation as 'the collective effect of service performance, which determines the degree of satisfaction of a user of the service'. Thus, the real degree of quality developed should be evaluated relatively to the achieved user satisfaction. But, how can we quantify user satisfaction?

This is an issue deeply analysed by enterprise quality management and social sciences, resulting in the development of different models and measurement methods. QoE-related research has mainly been aimed at developing a technical approach of perceived quality assessment for each specific service discretely (i.e., video, voice or web). Nevertheless, there have been some efforts to provide an overall vision of different (subjective/objective) aspects of QoS. For example in Gbaguidi et al. (1997), the authors proposed a QoS management architecture focused on end-users. They added an extension to the OSI reference model, by including a new level called 'end-user level'. However, they did not provide mechanisms for mapping the relationships between layers nor analytical instruments for extracting valuable information from the theoretical model. Corrie et al. (2003) and Patrick et al. (2004) applied the concept of QoE, originally defined by Alben (1996) to collaborative environments as 'the characteristics of the sensations, perceptions, and opinions of people as they interact with their environments'. Although they tried to consider all possible factors that may have an impact on users, their study was too focused on collaborative environments and the concept of session. They also proposed in Bauer and Patrick (2004) an extension of the OSI model which would include three additional layers: 'display', 'human performance' and 'human need'. This new OSI + HCI (human computer interaction) model is aimed at providing 'a consistent language' in order to map quality requirement in each level. But, once more, it was an abstract model without the tools to identify the dependencies between layers.

Finally, probably the most important initiative related to QoE was ITU-T G.1000 Recommendation in ITU-T (2004). This recommendation brought together previous work around the relationships between quality and network performance in other Recommendations such as E.800, I.350 or Y.1540 and represented them in a matrix that included both objective-subjective factors and both technical and non-technical aspects of the services (e.g., billing or customer support). However, there were no practical methodologies for applying the model and analysing the results.

3 The proposed integrated QoE framework

The model we propose tries to solve the mismatch between user-satisfaction and traditional objective QoS studies. The basis of the general model was first presented in Liberal et al. (2005). The model had a matrix structure similar to quality function deployment (QFD) (see Akao, 1990) quality methodology and ITU-T G.1000

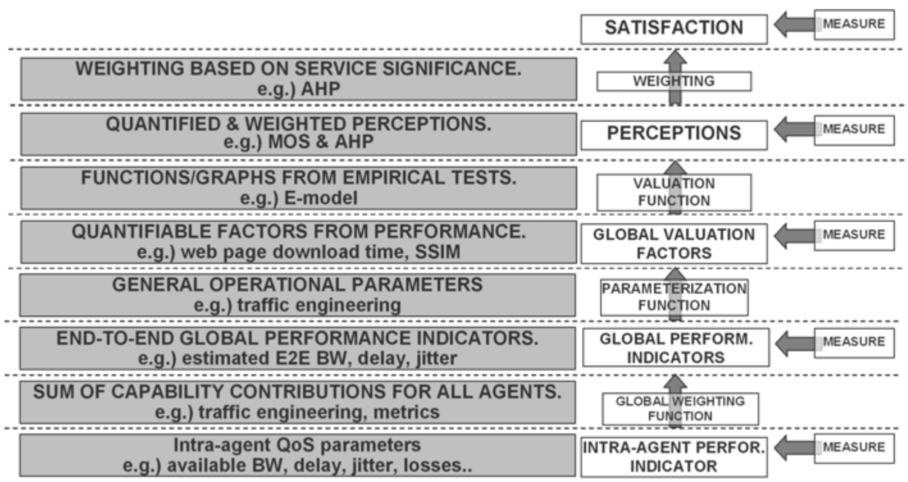
Recommendation (ITU-T, 2004), that allowed us to display the complex relationships between the agents involved in the audio-visual distribution chain and user perceptions for different services. However, in this study we have used a simplified and more practical version of the model, considering only those elements suitable for analysing the contribution of different agents in the telecommunication service provision to final QoE and the associated calculus process:

- First of all, we must identify those services under study and the relative importance of each one of them for our average user. The relative weights of each service will be estimated with the analytic hierarchy process (AHP) multicriteria decision tool defined in Saaty (1980) as done in Bodin et al. (2005) and Ghinea and Magoulas (2001).
- Then, for each service, we must identify those 'perceptions' relevant for gauging final users' QoE. Here, we call perceptions to those aspects related to quality of service that have an impact on users-satisfaction when accessing a specific service (e.g., interactivity, file downloading speed, audio or video 'quality', reliability, etc...).
- Each perception of quality will depend on several factors, both subjective ones (such as content characteristics, user expectations, prior experiences or their willingness to pay) and others related to objectively measurable performance. The latter are suitable to be managed by network administrators. These end-to-end objectively measurable factors that reflect the network performance and have a direct impact on user-perception are called global valuation factors (GVFs). GVFs are not traditional QoS parameters but are related to their effects [e.g., web downloading time (DT) or multimedia objective metrics are not directly QoS parameters such as throughput, delay and jitter]. Most of the times, the relationship between a single QoE value and GVF will be estimated by a utility function derived from QoE estimating techniques that map QoE vs. NQoS [see examples of utility functions in Fiedler et al. (2005) and Hoßfeld et al. (2006)].
- Sometimes, we want to inspect the contribution of a particular agent to end-to-end performance and, therefore, to final user-perception. That is why we have included locally (intra-agent) measured QoS parameters into the calculus process. End-to-end QoS parameters can be estimated from these per-agent parameters with simulation tools, traffic engineering techniques or using simple metrics as in Alkahtani et al. (2002) (e.g., adding intermediate delays to calculate e2e delay). From end-to-end QoS values, we will be able to calculate objective GVF by traffic engineering methods or/and protocol analysis.

Then, the overall calculus process is depicted in Figure 1: From local QoS statistics e2e NP indicators can be estimated. From these objective parameters, the global effect into the service (such as web page DT, voice and video specific performance metrics – SSIM – etc...) will be calculated. Finally, these GVFs provide the input for the utility functions to provide an estimation [quantified in a mean opinion score (MOS) scale] of user satisfaction regarding this service and AHP will weigh the contribution of each service into overall user satisfaction. Furthermore, any of the intermediate elements can be either calculated through estimation functions from lower levels indicators or we can use real/simulation measurements or users surveys in order not to cope with the complexity of these functions.

In order to feed the model, we need advanced specific methods for mapping NQoS parameters and estimated QoE. In the following subsections, we will review and summarise proposed methods for every service considered.

Figure 1 Estimated satisfaction calculus process



4 QoE vs. NQoS mapping methods of the proposed framework

4.1 Modelling/predicting web QoE

Most of the studies of perceived quality in web services concluded that the most important quality metric in web browsing, affected by network performance, is associated to page DT (e.g., Olshefski and Nieh, 2006) also called 'web latency' or 'web lag'. This conclusion is also statistically obtained through correlation and analysis of variance (ANOVA) between group tests in Saliba et al. (2005) and Zviran et al. (2006). Regarding DT impact on satisfaction, there are no unanimous criteria for modelling the relationship between DT and user perception. There have been identified several satisfaction thresholds (see Muntean and McManis, 2004; Woolley, 2000), so that a bunch of '# seconds rules' have appeared. For example, Nah (2004) collects another five different web DT tolerating thresholds and carries out a series of empirical investigations in order to provide their own values (2 and 15 seconds).

Then, although there are no standardised values, most of the authors agree that there are both maximum and minimum values for web usage perception and that, beyond these points, any improvement in performance does not have any impact on user-satisfaction. These thresholds may vary along the sessions, due to prior experiences of the users or the way information appears (e.g., with incremental or non-incremental image loading techniques) [see Bhatti et al. (2000) and Chung and Zhao (2004) for more details].

We have taken into consideration the results presented in Johnson (1998), Okamoto and Hayashi (2002) and ITU-T (2006), who analyse the users perception about web browsing by means of MOS questionnaires. As a result, the perception is approximated as a logarithmic expression [this is the simplified version, see ITU-T (2006) in order to consider other parameters such as 'time for the first response visible'].

$$MOS = 6 - \log_2 (DT) | 1 < MOS < 5 \quad (1)$$

The truncated logarithmic shape of the utility function fits the characteristics of user perception in terms of DT, since it provides a first stage (with very short DTs) with saturated maximum satisfaction, a second stage where satisfaction decreases and a final stage of saturated minimum satisfaction. In fact, logarithmic functions have been usually considered as a typical utility or cost function in studies related to minimising cost (Lee, 1999) or analysing users' willingness to pay (see Yamori and Tanaka, 2004) both for web and other types of services.

4.2 Modelling/predicting VoIP QoE

For VoIP applications, QoE modelling/predicting can be either on listening-only voice quality or on conversational voice quality which takes into account interactivity. In some previous work (see Sun and Ifeachor, 2006), we have demonstrated how to derive conversational MOS model from end-to-end packet loss and delay based on a combined ITU-T PESQ and e-model structure. We have followed a similar route to develop QoE model (in terms of MOS) based on MOS listening quality objective [MOS-LQO, see ITU-T (2003)], which is closer to subjective MOS score when compared to MOS score obtained from PESQ, referred to as MOS (PESQ). In the paper, we refer the MOS-LQO value obtained from PESQ mapping as MOS (PESQ-LQO), or abbreviated as PESQ-LQO. We develop models for four modern codecs (i.e., G.729, G.723.1, AMR and iLBC) under different sender bit rate conditions.

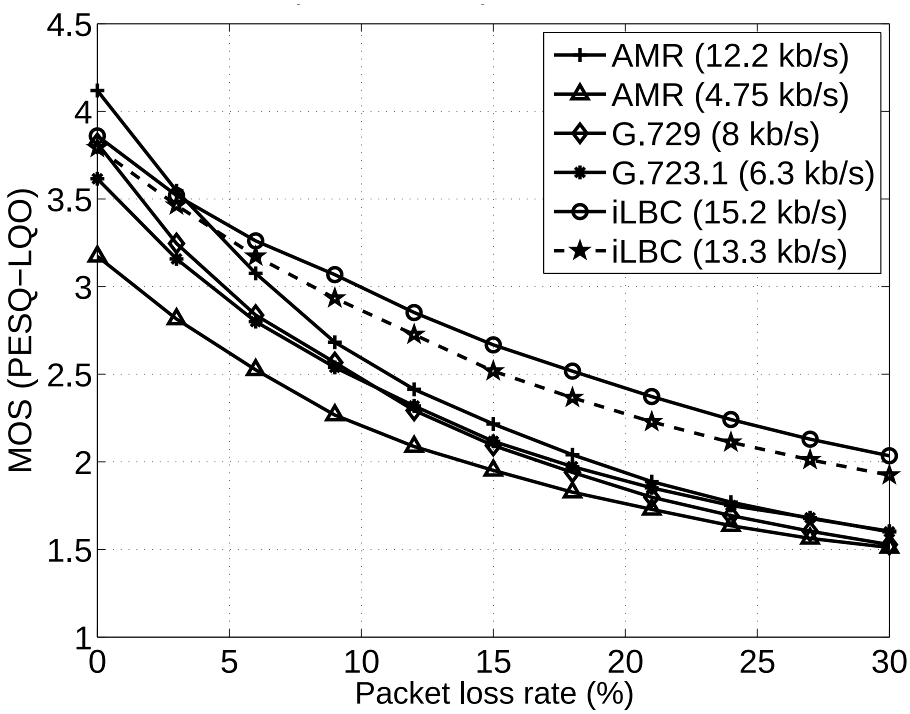
A VoIP simulation system is built up to simulate a VoIP flow, which includes encoder, packet loss simulator and decoder. The reference speech is taken from the ITU-T dataset. Packet loss is generated from 0% to 30%, in an incremental step of 3% and Bernoulli loss model is used for simplicity. Except G.729 with each packet containing two speech frames (corresponding to 20 ms speech per packet, with packet payload size of 20 bytes), all other codecs contain one speech frame per packet (for AMR/iLBC: 20 ms speech per packet with different payload size for different mode of the codec; for G.723.1: 30 ms speech per packet). All four codecs have internal packet loss concealment algorithms. No external packet loss concealment and/or recovery mechanisms are considered in the paper. ITU-T PESQ (ITU-T, 2001) is used for evaluating end-to-end voice quality by comparing the reference and the degraded speech samples. For each speech sample in the dataset for British English, a MOS (PESQ) score is obtained by averaging over 30 different packet loss locations (via using different random seed setting) in order to remove the influence of packet loss location. Further, the MOS score for a packet loss is obtained by averaging over all male and female speech samples (three of eight males and eight females), respectively. We notice that the average MOS for female is lower than that for male samples at all the test points. The value differences are between 0.01 to 0.29. The higher the packet loss rate, the larger the gap between MOS scores for male and female samples. In order to have a general model for each codec, we average the MOS score from both male and female samples. At the end, the overall MOS (PESQ) is mapped to MOS (PESQ-LQO) according to the following mapping function in ITU-T (2003):

$$y = 0.999 + \frac{4.999 - 0.999}{1 + e^{-1.4945x + 4.6607}}$$

where x and y represent MOS from PESQ and PESQ-LQO, respectively. MOS (PESQ-LQO) is regarded closer to subjective MOS score when compared to MOS (PESQ), as the MOS (PESQ-LQO) value is in the range of 1 to 5 (similar to the range for subjective MOS test), whereas, the MOS (PESQ) score is in the range of –0.5 to +4.5.

The results of MOS (PESQ-LQO) versus packet loss rate for different codecs are shown in Figure 2. It can be seen that iLBC (15.2 kb/s, 20 ms frame length) has the best packet loss robustness feature in all the codecs compared. AMR (4.75 kb/s) has the lowest voice quality no matter with or without packet loss. There is no obvious linear relationship between voice quality and codec's sender bit rate.

Figure 2 MOS versus packet loss rate for different codecs



The relationship between the MOS (PESQ-LQO) versus packet loss rate can be converted to the equipment impairment factor I_e (which represents effects of equipment such as

VoIP systems and codecs on the speech signal), versus packet loss rate via the following equations (see Sun and Ifeachor (2006) for more details). The equipment impairment factor (I_e) is an important element in the commonly used e-model described in ITU-T (2005) to reflect impairments from packet loss, jitter and codec in VoIP applications:

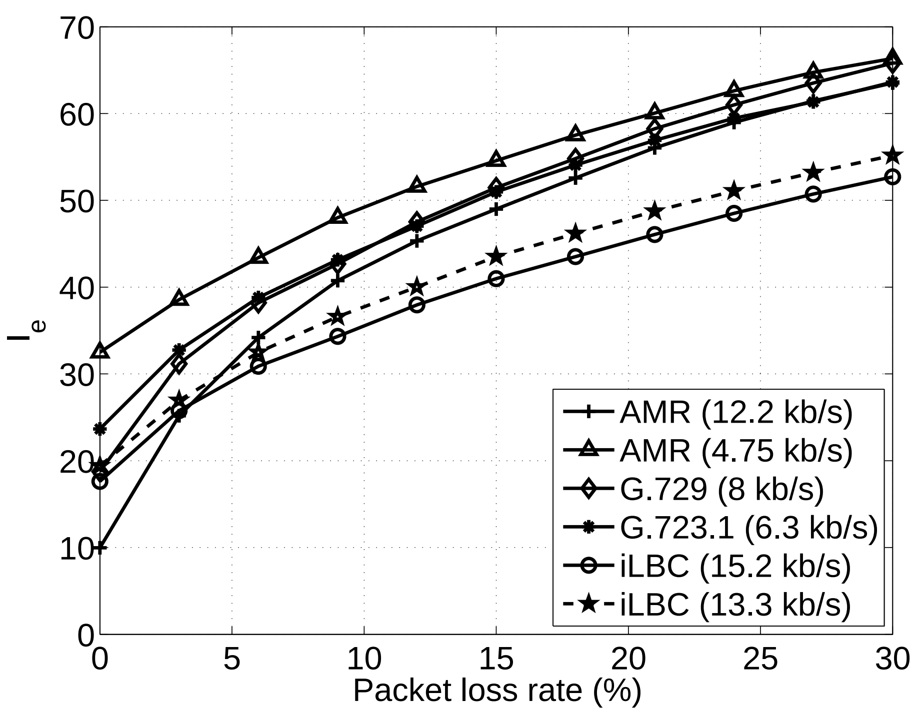
$$R = 3.026 \cdot MOS^3 - 25.314 \cdot MOS^2 + 87.060 \cdot MOS - 57.336$$
$$I_e = R_o - R = 93.2 - R$$

The derived curves for I_e versus packet loss rate are shown in Figure 3. A logarithmic fitting function of I_e versus packet loss rate $(\rho,$ in percentage) for each codec can be derived in the following form:

$$I_e = a \ln(1 + b \cdot \rho) + c$$

The fitting parameters for the six selected codec cases (including different bit rates) can be obtained by non-linear least square curve fitting and are shown in Table 1. The R^2 factor for the goodness-of-fit is also listed.

Figure 3 I_e versus packet loss rate ρ



Note: Based on PESQ-LQO

Table 1 Fitting parameters for I_e versus packet loss for different codecs

Parameters	*AMR (12.2 kb/s)*	*AMR (4.75 kb/s)*	*G.729 (8 kb/s)*	*G.723.1 (6.3 kb/s)*	*iLB (15.2 kb/s)*	*iLBC (13.3 kb/s)*
a	22.9789	26.4596	24.6019	24.2290	21.9999	23.0987
b	0.3054	0.0879	0.1844	0.1375	0.1245	0.1214
c	10.0653	32.4215	19.2603	23.9083	18.0696	19.5654
R^2 factor	0.9997	0.9998	0.9986	0.9995	0.9986	0.9998

The delay impairment, I_d, representing all impairments due to delay of voice signals such as talker/listener echo and absolute delay can be derived from end-to-end delay (d), given in ms, using the following equation (see Cole and Rosenbluth, 2001). I_d is another important element in the e-model in ITU-T (2005).

$$I_d = 0.024 \cdot d + 0.11 \cdot (d - 177.3) \cdot H(d - 177.3), \quad \text{where} \begin{cases} H(x) = 0, & \text{if } x < 0 \\ H(x) = 1, & \text{if } x \geq 0 \end{cases}$$

Based on I_e and I_d, the R- factor from the e-model can be derived by:

$$R = R_0 - I_d - I_e = 93.2 - I_d - I_e$$

From the R- factor, the MOS value can be obtained by:

$$MOS = \begin{cases} 1 & \text{for} \quad R \leq 0 \\ 1 + 0.035 \cdot R + R \cdot (R - 60) \cdot (100 - R) \cdot 7 \cdot 10^{-6} & \text{for} \quad 0 < R < 100 \\ 4.5 & \text{for} \quad R \geq 100 \end{cases}$$

These models can be directly used for predicting voice quality under different network conditions and for QoS control to achieve best trade-off between packet loss and delay. It needs to be mentioned that when external packet loss concealment or recovery mechanisms are used, the voice quality (in terms of MOS score) of a VoIP system can be improved, especially in lower packet loss conditions. In these cases, new curves of MOS versus packet loss can be developed and new functions can be derived in the methods/procedures mentioned above.

4.3 Modelling/predicting video QoE

In digital video encoding, the block discrete cosine transformation (BDCT) is exploited, since it exhibits very good energy compaction and de-correlation properties. In this paper, we use the following conventions for video sequences: Every real N × N frame f is treated as a N^2x1 vector in the space R^{N^2} by lexicographic ordering by either rows or columns. The DCT is considered as a linear transform from $R^{N^2} \rightarrow R^{N^2}$. Thus, for a typical frame f, we can write:

$$F = Bf$$

The high compression during the MPEG-related encoding process is (among other procedures) based on the quantisation of the DCT coefficients, which in turn results in loss of high frequency coefficients. Thus, an error based framework in the luminance domain Δf_Y between the original and the decoded frame will quantify QoE degradation per frame due to the encoding and quantisation process. A perceived quality metric, which provides very reliable assessment of the video quality, based on this error-based framework, is the structural similarity (SSIM) metric. The SSIM is a full reference (FR) metric for measuring the SSIM between two image/video sequences, exploiting the general principle that the main function of the human visual system is the extraction of structural information from the viewing field. Thus, if x and y depicts two video signals, then SSIM is defined as:

$$SSIM(x,y) = \frac{\left(2\mu_x\mu_y + C_1\right)\left(2\sigma_{xy} + C_2\right)}{\left(\mu_x^2 + \mu_y^2 + C_1\right)\left(\sigma_x^2 + \sigma_y^2 + C_2\right)}$$

where μ_x, μ_y are the mean of x and $y, \sigma_x, \sigma_y, \sigma_{xy}$ are the variances of x, y and the covariance of x and $y,$ respectively. The constants C_1 and C_2 are defined as:

$$C_1 = \left(K_1 \cdot L\right)^2; C_2 = \left(K_2 \cdot L\right)^2$$

where L is the dynamic pixel range and $K_1 = 0.01$ and $K_2 = 0.03,$ respectively (see Wang et al., 2004).

The concept of the mean SSIM for the whole video duration can be exploited for deriving a single perceived quality measurement, as it has been already done on the relevant literature. Towards this, a set of reference and real video clips (i.e., movie trailers) was used. Each movie trailer (and each reference clip) was transcoded from its original H.264 format with Hi-Def resolution (i.e., 720 p) (and PAL resolution) to ISO H.264 baseline profile, at different VBR bit rates. For each corresponding bit rate, a different H.264 compliant file with common interface format (CIF) resolution (352 × 288) was created. The frame rate was set at 25 frames per second (fps) during the transcoding process of all the test signals.

Each H.264 video clip was then used as input in the SSIM estimation algorithm. From the resulting SSIM vs. time graph, the average <QoE> value of each clip was calculated in terms of mean SSIM. This experimental procedure was repeated for each video clip in CIF resolution. The results of these experiments are depicted in Figure 4. Referring to the curves, it can be observed that the shape of each curve depends on the S-T activity level of the video content.

Moreover, each $<QoE>_{SSIM}$ vs. bit rate curve can be successfully described by a logarithmic function of the general form:

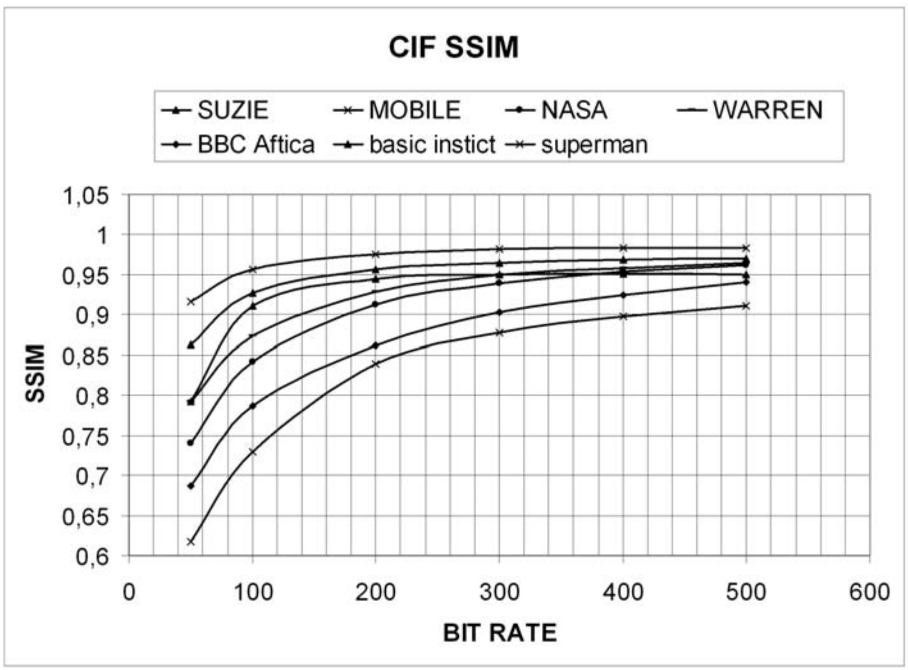
$$< QoE >_{SSIM} = C_1 \ln(x) + C_2 \tag{2}$$

where C_1 and C_2 are constants strongly related to the S-T activity level of the content and x is the bit rate in kbps. Table 2 depicts the corresponding logarithmic functions for the test signals of Figure 4 along with their R^2 factor, which denotes the fitting efficiency of the theoretical graph to the experimental one.

Table 2 Fitting parameters and R^2 for the videos of Figure 4

Test signal	*Logarithmic function*	*R^2 factor*
Mobile&Calendar	$0.1295 \cdot \ln(x) + 0.1274$	0.9759
Imax-sea	$0.0563 \cdot \ln(x) + 0.6411$	0.9514
Warren	$0.0738 \cdot \ln(x) + 0.5210$	0.9528
Basic Instinct	$0.0631 \cdot \ln(x) + 0.5829$	0.7781
Suzie	$0.0443 \cdot \ln(x) + 0.7075$	0.8901
Imax-Nasa	$0.0950 \cdot \ln(x) + 0.3892$	0.9595
BBC – Africa	$0.1098 \cdot \ln(x) + 0.2702$	0.9875
Superman Returns	$0.0282 \cdot \ln(x) + 0.8167$	0.8859

Figure 4 The $< QoE >_{SSIM}$ vs. bit rate curves for various test signals



Based on the aforementioned analysis, we can describe the derived $<QoE>_{SSIM}$ vs. bit rate curve of each test signal with N total frames, which is encoded at bit rate n (in the range from $BitRate_{\min}$ to $BitRate_{\max}$) as a set C_{S-T}. In this set, each element F_n is a triplet, consisting of the $<QoE>_{SSIM}$ of the specific bit rate and the constants C_1 and C_2, which are derived by the analytical logarithmic expression of Table 2:

$$C_{S-T} \triangleq \left\{ m : \left(\frac{1}{N} \sum_{i=1}^{N} SSIM\left(f_i\right), C_1, C_2 \right)_n = F_n, n \in \left[BitRate_{\min}, BitRate_{\max}\right] \right\}$$

Thus, deriving the sets C_{S-T} for various contents, ranging from static to very high S-T ones, a reference hyper set RS, containing various C_{S-T} sets for specific spatiotemporal levels can be deduced: $RS = \left\{ C_{S-T_{Low}}, \ldots, C_{S-T_{High}} \right\}$. Hence, consider an unknown video clip, which is uncompressed and we want to predict its corresponding C_{S-T} set that better describes its perceived quality vs. bit rate curve before the encoding process. Then, we define for all the sets C_{S-T} the absolute difference value (ADV) between the first C_{S-T} triplet element (i.e., the <QoE>$_{SSIM}$ at a specific encoding $BitRate_i$) and the experimental measurement of the average $SSIM$ for the test signal at the encoding bit rate n, for which all the reference sets C_{S-T} have been derived:

$$ADV = \left| F_{BitRate_i} : \left(\frac{1}{N} \sum_{i=1}^{N} SSIM\left(f_i\right) \right) - F'_{BitRate_1} : \left(\sum_{i=1}^{N} SSIM\left(f'_i\right) \right) \right|$$

Due to the fact that the additive property is valid, it is concluded that when the ADV between reference $F_{BitRate_i}$ and experimental $F_{BitRate_i}$ average $SSIM$ is minimum, then the set C_{S-T}, which contains the triplet element that minimises the ADV, describes better the specific video. The case of perceived quality degradation of the video during the streaming process due to network congestion is not examined, since we consider that the defined encoding quality is not degraded during the service transmission and the requested encoded-quality is delivered to the end-user.

4.4 Modelling/predicting online games QoE

Several authors have analysed the impact of the delay on the players' satisfaction. This impact may vary according to game-dependant factors [such as, for example the kind of weapon used, see Svoboda and Rupp (2005)]. Most of them simply establish a maximum tolerable delay threshold that users can stand. In order to provide a detailed relationship, we have used the minimum and maximum delay thresholds as parameters to build up a family of utility functions as in Johnson (1998) and Richards et al. (1998) with the expressions in the following equations. As we have shown in Section 4.1 regarding web QoE, truncated logarithmic functions are commonly used as utility functions and consistent with subjective experiments. In this case, we use commonly accepted thresholds in order to customise generic curves to utility function for online games QoE.

$$s\left(x\right) = 5 - 4 \cdot a \cdot \ln\left(b \cdot x + c\right) \tag{3}$$

where

$$a = \frac{1}{p-10} \qquad b = \frac{\left(\exp\left(\frac{1}{a}\right)-1\right)}{\left(S_{\max}-S_{\min}\right)} \qquad c = \frac{\left(S_{\max}-S_{\min}\cdot\exp\left(\frac{1}{a}\right)\right)}{\left(S_{\max}-S_{\min}\right)}$$

$S_{\max}$ is the maximum tolerable delay threshold

$S_{\min}$ is the minimum noticeable delay threshold

p is often called 'sensitivity factor', since it controls the sensitivity of satisfaction to variations in x.

With a minimum noticeable delay of 20 ms, and maximum tolerable delay threshold of 150 ms, a family of utility functions is obtained (for different p). Depending on the value of the p parameter, we can choose between more or less 'tolerant' users in a similar way as the classification of complainers and optimistics in Dick et al. (2005).

5 Application of the model to troubleshoot multimedia services provisioning

One of the main purposes of the model and calculus proposed in this paper is the provision of useful information to managers on the impact of network performance to perceived quality or QoE. By using this calculus process, it is possible to obtain an estimation of end-user satisfaction. Furthermore, once the proposed model has been built and evaluated, analytical methods can be further used in order to extract valuable information for identifying both subjective (i.e., insensitivity of the perception to variations of the performance beyond certain thresholds) and objective (i.e., typical network performance related) bottlenecks, troubleshooting network issues or supporting decisions regarding selection among content or access providers.

5.1 Case study

In this section, a case study is presented for evaluating purposes of the proposed model, which assesses the perception of quality towards different multimedia services by different user profiles. The final goal is spotting the most important factors that affect the end-users. The network scenario considers a typical internet access, so the agents considered in the model are the user platform, the access network, the ISP (formed by intra-ISP connectivity and external connectivity), the inter-ISPs links and the content provider. Although the model allows the inclusion of pure subjective perceptions through MOS surveys (e.g., related to price, customer care system, etc...), we will analyse just the network performance-related perceptions only, in order to show the capability of the model to find those elements responsible for QoE degradation due to technical issues.

The relative importance of services for different users will depend on users preferences and usage habits. Furthermore, there are clear differences in users preferences between gender, age, geographical distribution, or occupation (see e.g., Fallows (2005) and Eurostat (2006) for in-depth studies) resulting in high market segmentation. When applying the model to any particular real scenario, the first stage will consist on carrying out a survey or estimating user's preferences using the AHP methodology.

The AHP methods provide us with a method for evaluating the impact of the composition of different services into users global satisfaction as a weighted sum. In this case, in order to evaluate the preferences of our 'hypothetical average user', we use a questionnaire and AHP to compare relative weights of considered services (namely web surfing, online games, VoD and VoIP). The application of AHP methodology comprises two steps. First, the user carries out a pairwise comparison between each services pair and scores them from 1 to 9 (or reciprocals) depending on how much more important one is as compared to the other.

Table 3 AHP general matrix

	Web	*Games*	*VoD*	*VoIP*
Web	1	5	1/2	1
Games	1/5	1	1/4	1/4
VoD	2	4	1	1/2
VoIP	1	4	2	1

Then, the weighted sum coefficients w_j for the composition of final satisfaction, according to AHP method are calculated from the a_{ij} coefficients as follows:

$$w_i = \frac{\left(\prod_{j=1}^{p} a_{ij}\right)^{\frac{1}{p}}}{\sum_{i=1}^{p}\left(\prod_{j=1}^{p} a_{ij}\right)^{\frac{1}{p}}} \quad i = 1, 2, \ldots p$$

Resulting, for the hypothetical average user in Table 3:

$w_{web} = 0.2660, w_{games} = 0.0684, w_{VoD} = 0.3067, w_{VoIP} = 0.3590$

In order to probe the consistency of the AHP process, the consistency ratio is computed resulting in $CR = 0.0793$. Since $CR < 0.1$, the evaluation is consistent according to AHP methodology [see Saaty (1980) for the mathematical details of the CR].

In order to simulate the behaviour and values of different interactions and intermediate elements of the model, we have taken into account the following constraints/simplifications:

- Available throughput, delay and losses have been considered as intra-agent QoS parameters.
- Simple multiplicative, additive and concave metrics have been used to obtain e2e values from intra-agent QoS parameters as in Alkahtani et al. (2002).
- The estimation of GVFs from e2e QoS parameters has been carried out as follows:

- An HTTP analysis was performed in order to get an expression for the web pages DT from e2e-available throughput and delay. This analysis assumes persistent connections with pipelining and TCP-effective throughput $TCP_{goodput}$. This is not the most realistic approximation, but it provides us with an upperbound limit for user-satisfaction, since it supposes full available bandwidth utilisation Thus, DT can be estimated, for a webpage with N web objects embedded (e.g., images, sounds, scripts, css,...) of size S_i :

$$DT = T_{DNS} + 2 \cdot RTT + \frac{S_{MAIN}}{TCP_{goodput}} + \sum_{i=1}^{N} \frac{S_i}{TCP_{goodput}} \quad (4)$$

- We have also assumed 'ping time' for online games equals RTT by considering processing time as negligible.

- The used valuation functions have been extracted from state-of-the-art and our own newly developed empirical studies collected in Section 2 with the following average values:
  - C_1 and C_2 factors for an average video ($C_1 = 0.0282$ and $C_2 = 0.8167$), with no sensitive network effect (so, considering adaptative codecs and encoding bit rate as the only effective factor for video)
  - G.729 codec ($a = 24.6019$; $b = 0.1844$; $c = 19.2603$) in VoIP
  - $p = 8$ for game quality evaluation.
- Simulation values:
  - ISP external connectivity varies from a throughput and delay range of (0–512 kbps) and (0–1000 ms) respectively
  - stationary delay, throughput and losses parameters for the rest of the agents.

The final results are shown in Table 4.

The model was fed with data under aforementioned constraints and simulated in MATLAB. For illustration purposes, the user satisfaction (in a MOS scale from 1 to 5) has been estimated in order to show the effect of different performance conditions of a particular service (web service in this case). In Figures 5 and 6, different number of N web objects for each web session (N = 1, and 15 respectively) are considered. Then, the estimated satisfaction versus only our ISP's external delay and throughput was plotted to simplify the figures. In fact, the simulation methodology itself is a kind of factorial experiments, since we are interested on evaluating the output effects by varying specific input variables of interest (in this case, the ISP's ones).

Analysing the resulting estimation, it is seen from Figure 5 that the network conditions have low impact on the web service-related satisfaction due to the small data-size of web pages and therefore the perceived QoE is good.

However, for N = 15, it is derived that the web service modifies the shape of the satisfaction, since for higher N values there is a stronger correlation with the available throughput.

As a result, a simple analysis of graphical information from the model can provide information about different contributions into overall satisfaction.

Table 4 Model application summary

	Global end users' satisfaction			
	AHP			
	Web	*Game*	*VoD*	*VoIP*
Perception	Browsing speed	Interactivity	Video quality	Audio quality
Valuation function	(1)	(3)	(2)	E-model
GVF	Download time (DT)	Ping time	SSIM	End-to-end downstream and upstream delay, available throughput and losses
Param. function	(4)	Ping time ~= RTT	NA	NA
e2e performance indicators	End-to-end downstream delay, end-to-end upstream delay, 'effective' download speed, webpage characteristics (number and sizes of the objects), total delay due to DNS queries	End-to-end downstream delay, end-to-end upstream delay	End-to-end available bit rate, resolution, frame rate	End-to-end downstream and upstream delay, available throughput and losses
Intra-agent performance indicators	Downstream delay for each agent, upstream delay for each agent, 'effective' download speed in an agent, web page characteristics (number and sizes of the objects)	Downstream delay for each agent, upstream delay for each agent	Bit rate, resolution, frame rate per agent	Downstream and upstream delay, available throughput and losses

Figure 5 Estimated satisfaction vs. ISP's external throughput and delay, $N = 1$

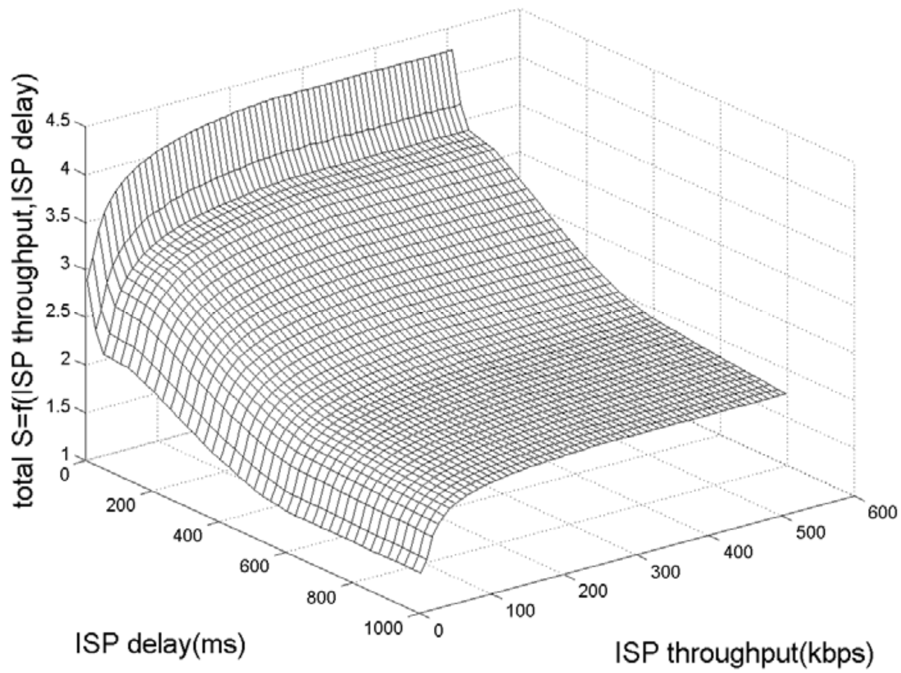
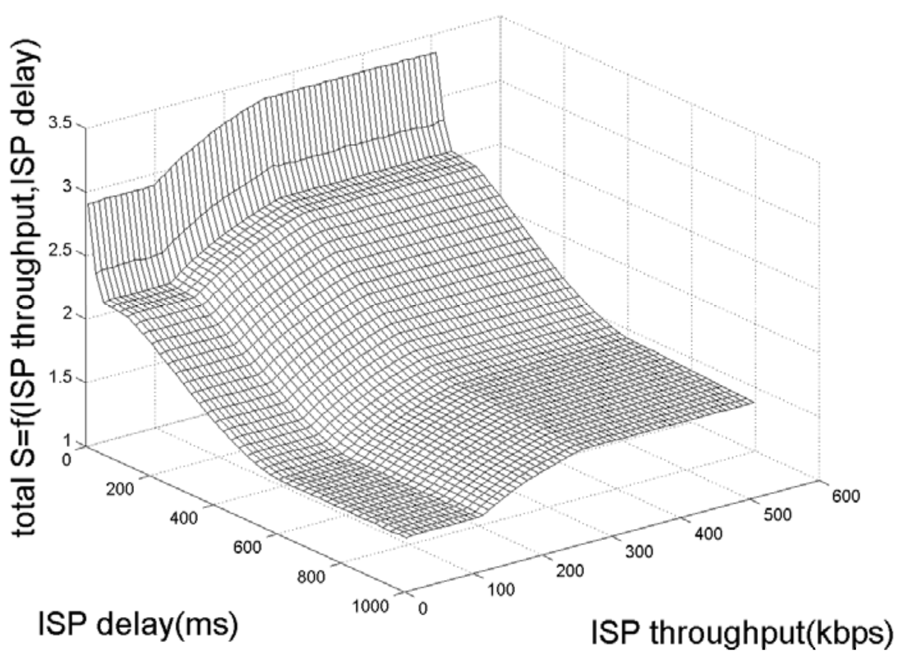


Figure 6 Estimated satisfaction vs. ISP's external throughput and delay, $N = 15$



Upon the estimation of average user's satisfaction, our aim consisted in analysing the responsibility of each one of the agents taking part in the telecommunication service provision. Towards this, the complex relationships between estimated satisfaction in a MOS scale versus different internal QoS parameters (throughput, delay and losses for user platform, access network, ISP's internal connectivity, ISP's external connectivity, inter-ISPs connectivity and the content provider) are modelled.

$$\begin{aligned} M\hat{O}S \; &= f(throughput_{UserPlat.}, delay_{UserPlat.}, losses_{UserPlat.}, \ldots, \\ &\quad throughput_{ContentProv.}, delay_{ContentProv.}, losses_{ContentProv.}) \end{aligned}$$

where $f(\cdot)$ is indeed the result of different interactions within the model and, therefore, with no analytical solution. The stepwise multiple linear regression method allows us to obtain the regression coefficients for a multiple linear model as:

$$\hat{y} = b_0 + b_1 \cdot x_1 + b_2 \cdot x_2 + \ldots + b_k \cdot x_k$$

so that

$$\begin{aligned} M\hat{O}S \approx\; & b_0 + b_1 \cdot throughput_{UserPlat.} + b_2 \cdot delay_{UserPlat.} + \\ & b_2 \cdot losses_{UserPlat.} + \ldots + b_{16} \cdot throughput_{ContentProv} \cdot + \\ & b_{17} \cdot delay_{ContentProv.} + b_{18} \cdot losses_{ContentProv.} \end{aligned}$$

So, in this case, we try to estimate the behaviour we have simulated with a multiple linear regression model, so that:

- our input variables $x_1, x_2, \ldots, x_k$ are traditional QoS parameters (available throughput, delay and losses) for user platform, access network, ISP's internal connectivity, ISP's external connectivity, inter-ISPs connectivity, and the content provider
- the output variable $\hat{y}$ represents the estimated QoE (e.g., MOS)
- b_k are the coefficients of the regression model and therefore quantify the relative importance of each input variable in the linear model.

However, Figures 5 and 6 do not show general linear behaviour along any axis for the whole range of values. In order to carry out this approximation, we will recalculate the multiple linear regression model for every single network state of ISP's external connectivity (so, in a small range where linearity can be assumed). The linearity approximation can be tested with residuals plot as done in Figure 7.

In order to properly compare these different inputs, which belong to different magnitudes and show different ranges of values, we have normalised each input by its variance using the so called Beta Coefficients for the multiple linear regression method obtaining:

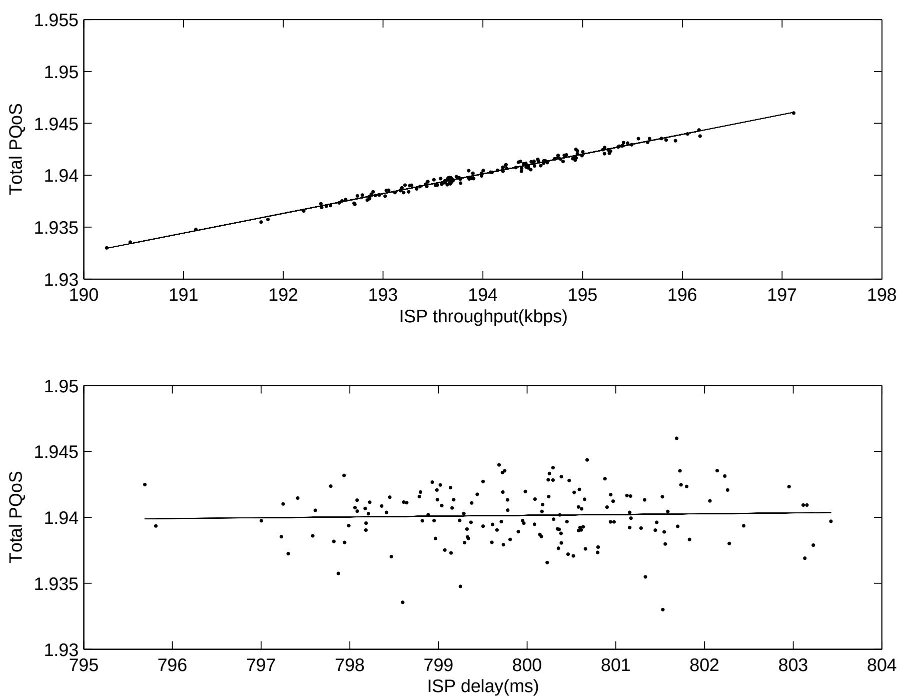
$$\hat{y}' = \beta_0 + \beta_1 \cdot x_1' + \beta 2 \cdot x_2' + \ldots + \beta_k \cdot x_k'$$

where

$$\hat{y}' = \frac{\hat{y}}{\sqrt{S_{yy}}}, x_i' = \frac{x_i}{\sqrt{S_{x_i x_i}}}, \beta_i = b_i \cdot \frac{\sqrt{S_{x_i x_i}}}{\sqrt{S_{yy}}}$$

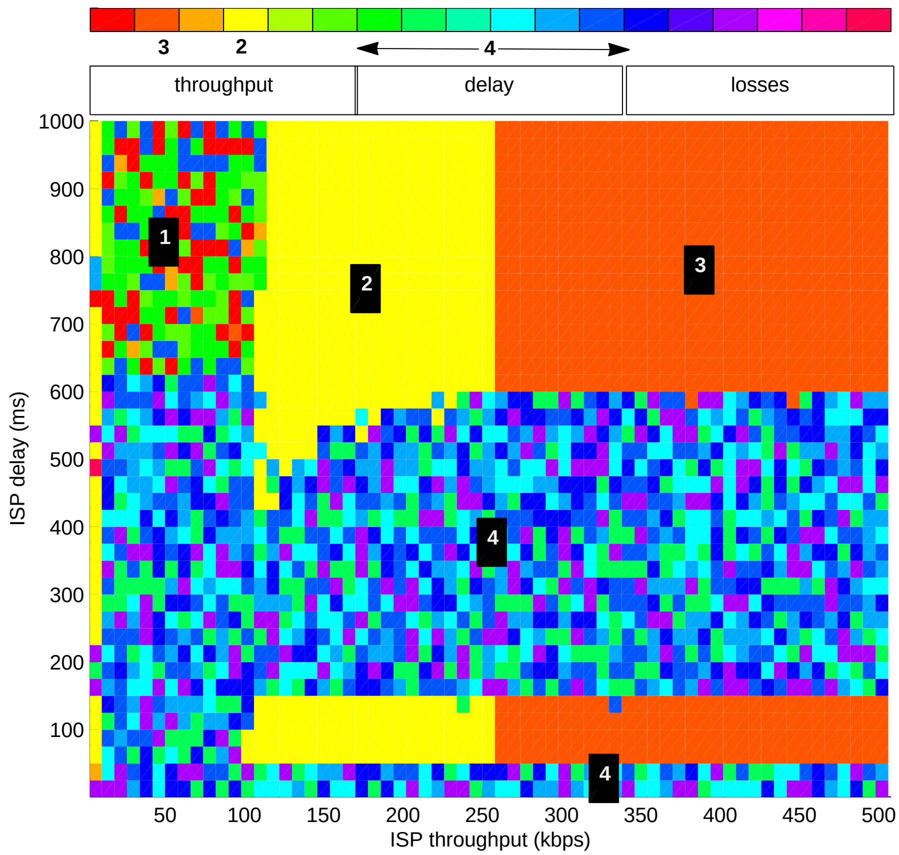
and $S_{x_i x_i}$ is the variance of the variable x_i. So, the new x_i' and $\hat{y}'$ variables are adimensional and the regression coefficients β_i do not depend on the magnitude of the variables.

Figure 7 Residuals plot for a single network state of ISP's external connectivity



Finally, the Beta coefficients of the regression model were calculated by using the stepwise multiple linear regression calculus process. In Figure 8, the input factor with highest coefficient is depicted for every network state of our ISP's external connectivity (throughput and delay axes in the figure). This analysis allows us to assess the relative importance of the QoS parameters of each agent considered under some circumstances of our ISP. Each greyscale resembles one of the input QoS parameters for every agent considered (user platform, access network, ISP's internal connectivity, ISP's external connectivity, inter-ISPs connectivity and content provider), which means 18 different colours (in a greyscale) for 18 input variables.

Figure 8 Main contributor to end user's satisfaction (see online version for colours)



Analysing Figure 8, we can find different zones:

1 no-dependence region

2 zone of predominant influence of ISP's external available throughput

3 zone of predominance of access network throughput

4 equivalent predominance of delay for all agents.

In Region 1, there is neither satisfaction variation nor predominance of any factor into users satisfaction. It is called a 'subjective bottleneck' since it is related to the 'bad quality' situation in the valuation functions of all services, due to low levels of throughput and high delays. Every NQoS improvement within this area will result in no satisfaction variation (in other words, it will be worthless). Therefore, any investment in technology should result in an improvement in QoS performance only if it moves the network state out of this region.

The second region corresponds to ISP external available throughput, whose importance is derived from the web and video services dependence on throughput.

There is a transition point for ISP's external available throughput = 256 kbps where there appears an objective bottleneck (due to access network 256 kbps bandwidth). This 'objective bottleneck' sets the threshold between Regions 2 and 3, where satisfaction is dominated by access network throughput limit. Objective bottlenecks are detected where there exists a change between two agent's predominance and it is related to concave metrics (e2e throughput calculated as the lowest throughput in the chain).

Finally, in Region 4 the predominance is shared between agents, due to the additive nature of delay and the contribution of agents to VoIP and online games services. In this region any improvement on the delay caused by any agent will have similar impact on user satisfaction.

6 Conclusions

This paper proposes a general QoE-based quality management model that provides methods for evaluating the relationships between user satisfaction and individual agents that take part in telecommunication service provision.

In order to test the model, we have proposed a case study: troubleshooting QoE in a typical scenario (internet access and the average user's service use). We have used both newly developed and widely accepted perception assessment methods for VoIP, video, web and online games services in order to provide a link between objective and subjective aspects of quality. In order to do so, the AHP multicriteria decision tool makes it possible to weight the relative importance of each service in the overall satisfaction for comparison purposes. Once we have built the model, we test its utility in order to extract valuable information beyond pure technical QoS parameters. So, by using stepwise multiple linear regression beta coefficients, we have been able to quantify the relative importance of each agent's contribution to the actual quality as perceived by users. Thus, we have identified both subjective and objective bottlenecks. The former is related to a saturation area where the user is insensitive to any performance enhancement, so that, in order to really have an impact into user satisfaction the improvement of the QoS must cross the borders of the area. The latter is related to traditional technical QoS bottlenecks where an improvement in a particular agent in the service delivery chain does not have an impact into e2e QoS levels. As an important conclusion of this analysis, we state that any improvement resulting in end-to-end objective QoS variation within this area is of limited usefulness. Finally, the model is even capable of detecting objective bottlenecks, those points where different agents interchange their role of most important element in service provision.

Acknowledgements

Part of the work in this paper has been performed within the framework of the ICT-214751 project ADAMANTIUM (European R&D FP7).

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